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ART UNIT	**PAPER NUMBER**
2654	

DATE MAILED: 10/23/2002

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	09/302,397	OZAWA, KAZUNORI
	Examiner	Art Unit
	Angela A. Armstrong	2654

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 02 August 2002 .

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-11 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-11 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

11)☐ The proposed drawing correction filed on _____ is: a)☐ approved b)☐ disapproved by the Examiner.

If approved, corrected drawings are required in reply to this Office action.

12)☐ The oath or declaration is objected to by the Examiner.

Priority under 35 U.S.C. §§ 119 and 120

13)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____ .

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

14)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e) (to a provisional application).

a) ☐ The translation of the foreign language provisional application has been received.

15)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. §§ 120 and/or 121.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)	4) ☐ Interview Summary (PTO-413) Paper No(s). _____ .
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)	5) ☐ Notice of Informal Patent Application (PTO-152)
3) ☐ Information Disclosure Statement(s) (PTO-1449) Paper No(s) _____ .	6) ☐ Other: .

DETAILED ACTION

Claim Rejections - 35 USC § 112

1. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

2. Claims 1-11 are rejected under 35 U.S.C. 112, first paragraph, as containing subject matter which was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention. Claims 1-11, as argued by applicant, include subject matter for processing pulses by using different pulse-shifting schemes depending upon whether a voice sound mode or n unvoiced sound mode is discriminated. Applicant has argued "Either way, a time-shifting scheme is employed".

3. Contrary to applicant's argument, the specification does not support a time shift of both voiced and unvoiced sound modes. Referring to the specification, page 26, lines 14-19, the time shift amount is indicated by $\delta(j)$. The time shift amount is indicated in the equations for minimizing distortions of unvoiced signal, as illustrated in equations 15, 17, 19, and 21. However, the shift amount is not indicated in the equations for minimizing distortions in the voiced mode, as illustrated in equations 11 and 16.

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

5. Claims 1-11 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kleijn et al (US Patent No. 5,704,003) in view of Swaminathan et al (US Patent No. 5,751,903) and Gershon et al (US Patent No. 5,657,418).

Regarding claims 1 and 6, Kleijn et al teaches a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3.

Additionally, Kleijn teaches an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;

Kleijn also discloses a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Gershon discloses a speech coder with the provision of gain information using multiple coding modes (col. 5, line 66 continuing to col. 6, line 13) and teaches that the lag parameter, which reflects the periodicity, is used to select a particular coding mode (col. 4, lines 16-19; 40-45; 48-53; col. 2, lines 28-31). Gershon et al teach that the system is useful in reducing speech coder data rates and maintaining or improving good speech quality (col.2, lines 7-11)

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Gershon et al, for the purpose of reducing speech coder data rates and maintaining or improving good speech quality, as suggested by Gershon et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of pulses and amplitudes and searches code vectors stored in the codebook and delays or shift amounts so as to output a combination of code vector and shift amount that minimizes distortion at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

In a similar field of endeavor, Swaminathan teaches a multi-mode CELP codec apparatus which implements a mode determining section, pulse codebooks, codebook searching and gain quantization, multiplexing spectrum parameters, codebook and quantization outputs for transmission to a decoder, a decoder which demultiplexes the transmitted spectrum parameters,

codebook and quantization outputs, determines modes, and reconstructs the sound source signal via synthesis (col. 5, lines 8-39), for the purpose of providing high-quality speech coding and decoding.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Swaminathan et al.

Regarding claims 2, 5, and 7, Kleijn discloses spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn disclose an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62 and a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Gershon discloses a speech coder with the provision of gain information using multiple coding modes (col. 5, line 66 continuing to col. 6, line 13) and teaches that the lag parameter, which reflects the periodicity, is used to select a particular coding mode (col. 4, lines 16-19; 40-45; 48-53; col. 2, lines 28-31). Gershon et al teach that the system

is useful in reducing speech coder data rates and maintaining or improving good speech quality (col.2, lines 7-11)

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Gershon et al, for the purpose of reducing speech coder data rates and maintaining or improving good speech quality, as suggested by Gershon et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discriminating section, and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

In a similar field of endeavor, Swaminathan teaches a multi-mode CELP codec apparatus which implements a mode determining section, pulse codebooks, codebook searching and gain quantization, multiplexing spectrum parameters, codebook and quantization outputs for transmission to a decoder, a decoder which demultiplexes the transmitted spectrum parameters, codebook and quantization outputs, determines modes, and reconstructs the sound source signal via synthesis (col. 5, lines 8-39), for the purpose of providing high-quality speech coding and decoding.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Swaminathan et al.

Regarding claims 3, 8 and 11, Kleijn discloses a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn discloses an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;

Kleijn discloses a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Gershon discloses a speech coder with the provision of gain information using multiple coding modes (col. 5, line 66 continuing to col. 6, line 13) and teaches that the lag parameter, which reflects the periodicity, is used to select a particular coding mode (col. 4, lines 16-19; 40-45; 48-53; col. 2, lines 28-31). Gershon et al teach that the system is useful in reducing speech coder data rates and maintaining or improving good speech quality (col.2, lines 7-11)

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode

is based on a past quantized gain, as taught by Gershon et al, for the purpose of reducing speech coder data rates and maintaining or improving good speech quality, as suggested by Gershon et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from the discriminating section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

In a similar field of endeavor, Swaminathan teaches a multi-mode CELP codec apparatus which implements a mode determining section, pulse codebooks, codebook searching and gain quantization, multiplexing spectrum parameters, codebook and quantization outputs for transmission to a decoder, a decoder which demultiplexes the transmitted spectrum parameters, codebook and quantization outputs, determines modes, and reconstructs the sound source signal via synthesis (col. 5, lines 8-39), for the purpose of providing high-quality speech coding and decoding.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Swaminathan et al.

Regarding claim 4, Kleijn teaches a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn teaches an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62; Discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Gershon discloses a speech coder with the provision of gain information using multiple coding modes (col. 5, line 66 continuing to col. 6, line 13) and teaches that the lag parameter, which reflects the periodicity, is used to select a particular coding mode (col. 4, lines 16-19; 40-45; 48-53; col. 2, lines 28-31). Gershon et al teach that the system is useful in reducing speech coder data rates and maintaining or improving good speech quality (col.2, lines 7-11)

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode

is based on a past quantized gain, as taught by Gershon et al, for the purpose of reducing speech coder data rates and maintaining or improving good speech quality, as suggested by Gershon et al.

Kleijn et al further teaches, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

In a similar field of endeavor, Swaminathan teaches a multi-mode CELP codec apparatus which implements a mode determining section, pulse codebooks, codebook searching and gain quantization, multiplexing spectrum parameters, codebook and quantization outputs for transmission to a decoder, a decoder which demultiplexes the transmitted spectrum parameters, codebook and quantization outputs, determines modes, and reconstructs the sound source signal via synthesis (col. 5, lines 8-39), for the purpose of providing high-quality speech coding and decoding.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Swaminathan et al.

Regarding claims 9-10, Kleijn, Gershon, and Swaminathan teach everything as claimed in claim 8. Additionally, at col.7, lines 2-26, Kleijn discloses determination of the time shift amount is based on a value that minimizes a certain criteria, which reads on "sound source quantization uses a position generated according to a predetermined rule as a pulse position when mode discrimination indicates a predetermined mode."

Response to Arguments

6. Applicant's arguments filed August 2, 2002 have been fully considered but they are not persuasive.

Regarding the 35 U.S.C. 112, first paragraph rejection, applicant argues the claims are supported by the specification. The Examiner disagrees and argues that the specification does not support a time shift for both voiced and unvoiced sound modes. As indicated in the previous Office Action and in the rejection above, the shift amount is only specifically disclosed in the specification for minimizing distortions of an unvoiced signal (equations 15, 17, 19 and 21). The shift amount is not indicated in the equations for minimizing distortions in the voiced mode (equations 11 and 16). Therefore, the rejection is proper and is maintained.

Regarding claim 1, applicant argues that the cited references fail to teach combinations of code vectors stored in the codebook and a plurality of shift amounts used to shift portions of the

pulses are searched so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech. The Examiner disagrees and argues that at col. 6, lines 21-61, Kleijn teaches a codebook search wherein the codebook vector and a time shifted signal are utilized to obtain an output signal which yields a target vector for a codebook search, which reads on "code vectors stored in the codebook and a plurality of shift amounts used to shift portions of the pulses are searched so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech."

Regarding claim 3, applicant argues that the cited references fail to teach combinations of code vectors stored in the codebook, a plurality of shift amounts used to shift portions of the pulses, and gain code vectors stored in the gain codebook are searched so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech. The Examiner disagrees and argues that at col. 6, lines 21-61, Kleijn teaches a codebook search wherein the codebook vector, a time shifted signal, and gain selection are utilized to obtain an output signal which yields a target vector for a codebook search, which reads on "code vectors stored in the codebook, a plurality of shift amounts used to shift portions of the pulses, and gain code vectors stored in the gain codebook are searched so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech."

In response to applicant's argument that the examiner's conclusion of obviousness is based upon improper hindsight reasoning, it must be recognized that any judgment on obviousness is in a sense necessarily a reconstruction based upon hindsight reasoning. But so long as it takes into account only knowledge which was within the level of ordinary skill at the time the claimed invention was made, and does not include knowledge gleaned only from the

applicant's disclosure, such a reconstruction is proper. See *In re McLaughlin*, 443 F.2d 1392, 170 USPQ 209 (CCPA 1971).

Conclusion

THIS ACTION IS MADE FINAL. Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Angela A. Armstrong whose telephone number is 703-308-6258. The examiner can normally be reached on Monday-Thursday 7:30-5:00 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Marsha Banks-Harold can be reached on (703) 305-4379. The fax phone numbers for the organization where this application or proceeding is assigned are 703-872-9314 for regular communications and 703-872-9314 for After Final communications.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the TC 2600 Customer Service Office whose telephone number is 703-306-0377.

Angela A. Armstrong
Examiner
Art Unit 2654

AAA
October 21, 2002

Marsha D Banks-Harold

MARSHA D. BANKS-HAROLD
SUPERVISORY PATENT EXAMINER
TECHNOLOGY CENTER 2600